RECEIVED 8-26-02 TL

In the Specification

Please amend the specification of this application as follows:

Rewrite the paragraph at page 4, lines 8 to 32 as follows:

B2

--In one embodiment of the present invention a system for reducing noise in an acoustical signal is provided. The system comprises a sampler for obtaining discrete samples of the acoustical signal, an analog to digital converter coupled to the sampler and operable to convert the analog discrete samples into a digitized sample, and a noise suppression circuit coupled to the analog to digital converter. The noise suppression circuit reduces noise by first receiving the analog discrete samples and then selecting a fixed number of samples. These samples are multiplied by a windowing function and the fast Fourier transform of the windowed samples is computed to yield transformed windowed signals. Half of the transformed windowed signals are selected and a power estimate of the transformed windowed signals is calculated. Next, a smoothed power estimate is calculated by smoothing the power estimate over time and a noise estimate is calculated. The noise estimate and the smoothed power estimate are used to calculate a gain function. A transformed speech signal is obtained by multiplying the gain function with the transformed windowed signal. Then, the inversed fast Fourier transform of the transformed speech signal is calculated to yield a sampled speech signal and the sampled speech signal is added to a portion of the speech signal of a previous frame.--

Rewrite the paragraph at page 7, line 19 to page 8, line 3 as follows:

B3

--FIGURE 2 illustrates a block diagram illustrating noise suppression unit 108 in accordance with the teaching of the present invention. Illustrated is a frame buffer 200 coupled to a windowing unit 202 which is coupled to a fast Fourier transfer module 204

which is then coupled to a noise reduction algorithm unit 206 which is then coupled to a inverse fast Fourier transfer module 208 which is finally coupled to a noise suppression frame buffer 210. In operation, frame buffer 200 partitions speech samples into frames of 32 samples. The sample frames are then sent to the windowing module 202 or an appropriate window function is applied. In one embodiment a Hanning window is applied. Fast Fourier transfer module 204 converts the frames to the frequency domain by using the well-known fast Fourier transform. Noise reduction unit 206 then invokes the main noise reduction algorithm. Noise reduction unit 206 takes the first 16 samples and computes the absolute value of the power of the sample. Then that power value is smoothed using the following equation.--

B3

Rewrite the paragraph at page 9, lines 7 to 8 as follows:

B4 --In step 302, the samples are multiplied by a Hanning window. A Hanning window is of the form--

Rewrite the paragraph at page 10, line 6 as follows:

B5 -- then $n^n(i)$ = downconst * $(n^{n-1}(i))$.--

Rewrite the paragraph at page 11, line 15 to page 12, line 2 as follows:

--In step 318, the inverse fast Fourier transfer is taken and in step 320, the sixteen computed values are added to the previous sixteen values. Then, in decision block 322 it is determined if there are any more already computed fast Fourier transition results awaiting calculation. If yes, the next 16 values are then calculated as before starting at step 308. If there are no more already calculated fast Fourier transfer value, decision box 324 is reached. In that box, it is determined it there is any more

B6